IN THE CLAIMS

1. (currently amended) An audio signal processing method comprising the steps of:

supplying an audio signal to each of a plurality of digital filters;

respectively supplying outputs from the plurality of digital filers to a plurality of speakers arranged in a speaker array to form a sound field;

setting a predetermined delay time in each of the plurality of digital filters so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filters and each of the plurality of speakers will coincide with each other; and

adjusting at least one amplitude characteristic of the plurality of digital filters such that the frequency response to the audio signal at the first point in the sound field is lower than the frequency response to the audio signal at a second point in the sound field~~so as to effect a spatially-localized low-pass filtering of the audio signal as output from the speakers, such that the audio signal at a second point in the sound field exhibits less higher frequency content than it would had the amplitude characteristic(s) not been adjusted, and such that the frequency content of the audio signal at the first point in the sound field remains substantially unchanged~~.

2. (previously presented) The audio signal processing method according to claim 1, wherein a sound wave from the speaker array is caused to reach at least one of the first and second points after it is reflected by a wall surface.

3. (previously presented) The audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the

plurality of digital filters is determined by calculation and set for each of the plurality of digital filters.

4. (original) The audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters.

5. (previously presented) The audio signal processing method according to claim 1, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal;

over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than a sampling period to provide a sample train, wherein the sample train is down-sampled to provide pulse-waveform data of the sampling period; and

factor data is set for a part to be delayed by the plurality of digital filters based on the pulse-waveform data.

6. (original) The audio signal processing method according to claim 5, wherein the audio signal is delayed by a part of the predetermined delay time, which is a multiple of the sampling period, by digital delay circuits which operate for the sampling period, while it is being delayed by the remainder of the predetermined delay time, which includes the decimal part by the digital filters.

7. (previously presented) The audio signal processing method according to claim 5, wherein:

an over-sampling period of the over-sampling operation is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

8. (previously presented) The audio signal processing method according to claim 7, wherein:

the pulse-waveform data to be delayed by a delay time which is m/N (m = 1 to N - 1) of the sampling period is pre-stored in a data base; and

pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

9. (previously presented) The audio signal processing method according to claim 5, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

10. (currently amended) An audio signal processor comprising a plurality of digital filters each supplied with an audio signal, wherein

each of the plurality of digital filters supplies an output signal to each of a plurality of speakers arranged in a speaker array to form a sound field;

each of the plurality of digital filters has a predetermined delay time so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filers and each of the plurality of speakers will coincide with each other; and

each of the plurality of digital filters has an amplitude characteristic such that the frequency response to the audio signal at the first point in the sound field is lower than the frequency response to the audio signal at a second point in the sound field~~so as to effect a spatially-localized low-pass filtering of the audio signal as output from the speakers, such that the audio signal at a second point in the sound field exhibits less higher frequency content than it would had the digital filters not had the amplitude characteristics, and such that the frequency content of the audio signal at the first point in the sound field remains substantially unchanged~~.

11. (previously presented) The audio signal processor according to claim 10, wherein a sound wave from the speaker array is caused to reach at least one of the first and second points after it is reflected by a wall surface.

12. (previously presented) The audio signal processor according to claim 10, wherein when forming the first and second points in the sound filter, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the plurality of digital filters.

13. (original) The audio signal processor according to claim 10, wherein when forming the first and second points in the sound field, a filter factor o each of the plurality of digital filters is read from a data based and set for each of the plurality of digital filters.

14. (previously presented) The audio signal processor according to claim 10, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal,

there is further provided a calculation circuit to calculate pulse-waveform data of the sampling period by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train; and

the pulse-waveform provided by the calculation circuit is set as a filter factor of each of the plurality of digital filters.

15. (previously presented) The audio signal processor according to claim 14, wherein:

an over-sampling period of the over-sampling in the calculation circuit is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, n/N is adopted as the decimal part.

16. (previously presented) The audio signal processor according to claim 14, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set synthetic-waveform data as a filter factor of each of the plurality of digital filters.

17. (previously presented) The audio signal processor according to claim 10, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal;

there is further provided a storing means for storing pulse-waveform data of the sampling period provided by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train; and

the pulse-waveform data stored in the storing means is taken out and set as a filter factor of each of the plurality of digital filters.

18. (previously presented) The audio signal processor according to claim 17, wherein:

an over-sampling period of the over-sampling is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

19. (previously presented) The audio signal processor according to claim 17, wherein:

a plurality of the pulse-waveform data corresponding to the decimal part is pre-stored in the storing means; and

pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

20. (previously presented) The audio signal processor according to claim 17, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set the pulse-waveform data as a filter factor of each of the plurality of digital filters.